



Avaya Solution & Interoperability Test Lab

Application Notes for LumenVox Automated Speech Recognizer, LumenVox Text-to-Speech Server and LumenVox Call Progress Analysis with Avaya Aura® Experience Portal – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate LumenVox Automated Speech Recognizer, LumenVox Text-to-Speech Server, and LumenVox Call Progress Analysis with Avaya Aura® Experience Portal.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

The objective of the compliance test was to validate interoperability of LumenVox Automated Speech Recognizer, LumenVox Text-to-Speech Server, and LumenVox Call Progress Analysis with Avaya Aura® Experience Portal.

LumenVox provides a complete set of speech recognition, text-to-speech technologies, and call progress analysis for use in interactive voice response (IVR) applications. The product set includes the LumenVox Automatic Speech Recognizer (ASR), Text-to-Speech (TTS) Server, and Call Progress Analysis (CPA). All products are used in conjunction with the LumenVox Media Server which provides an interface to Avaya Aura® Experience Portal using the Media Resource Control Protocol (MRCP). Additionally LumenVox Call Progress Analysis (CPA) leverages the strength of LumenVox Automated Speech Recognizer (ASR) by constantly listening for various tones, just as it would when performing speech recognition. These are compared to special acoustic models for matches, similar to our ASR's function. The result is more reliable and highly accurate message delivery.

2. General Test Approach and Test Results

The general test approach was to test various VoiceXML scripts that exercise various types of grammars in LumenVox ASR and TTS. A predefined set of VoiceXML scripts tested built-in grammars, menu grammars and Speech Recognition Grammar Specification (SRGS) grammars. Verification of CPA was performed using outbound calls for testing CPA capabilities.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability tests. Feature tests focused on the ability of LumenVox ASR and TTS to successfully exercise appropriate grammar and return expected results. LumenVox CPA focused on proper detection of answering party.

Serviceability testing focused on verifying the ability of the LumenVox ASR, TTS, and CPA server to recover from adverse conditions, such as restart, power failures and network disconnects.

2.2. Test Results

All test cases passed.

2.3. Support

To obtain technical support for LumenVox:

- **Web:** www.lumenvox.com/help/
- **Email:** support@lumenvox.com
- **Phone:** (858)707-7700

3. Reference Configuration

Following diagram shows the configuration used during the interoperability compliance test.

Reference configuration consisted of:

- Avaya Aura® Experience Portal
- Avaya S8300D Server running Avaya Aura® Communication Manager
- Avaya G450 Media Gateway
- Avaya Aura® Session Manager
- Avaya Aura® System Manager
- Avaya IP Telephones
- Application Server
- LumenVox Automated Speech Recognizer
- LumenVox Text-to-Speech Server
- LumenVox CPA

Note: Each of the LumenVox components are installed on a single server.

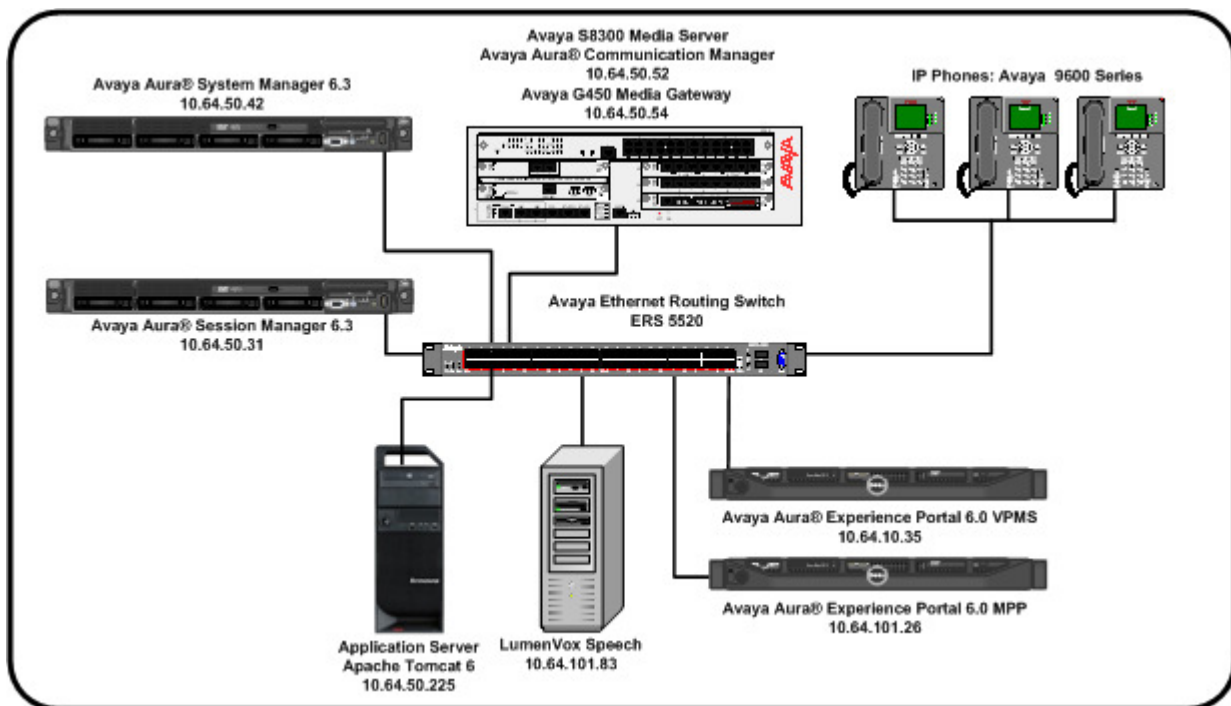


Figure 1: Reference Configuration

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration:

Equipment/Software	Release/Version
Avaya Aura® Experience Portal EPM running on HP Proliant DL360 G7	6.0.2.0.0501
Avaya Aura® Experience Portal MPP running on VM Ware Virtual Machine on Dell Blade Server	6.0.2.0.0501
Avaya G450 Media Gateway	32.26.0
Avaya S8300D Server running on Avaya Aura® Communication Manager	R016x.03.0.124.0
Avaya 9600 Series IP Telephones	H.323 3.2 SIP 6.2.2.17
Avaya Aura® Session Manager running on HP Proliant DL360 G7	6.3.2.0.632023
Avaya Aura® System Manager running on HP Proliant DL360 G7	6.3.0 –FP2
LumenVox components on a single VM Ware Virtual Machine running on Dell Blade Server: <ul style="list-style-type: none">• LumenVox Automated Speech Recognizer• LumenVox Text-to-Speech• LumenVox Call Progress Analysis	11.2.200
Tomcat Apache Web Server running on VM Ware Virtual Machine running on Dell Blade Server	6.0.29

5. Configure Avaya Aura® Experience Portal

Experience Portal is configured via the Experience Portal Manager (EPM) web interface. To access the web interface, enter <http://<ip-addr>/> as the URL in a web browser, where <ip-addr> is the IP address of the EPM. Log in using the admin user. (Not Shown)

AVAYA

Welcome, admin
Last logged in today at 11:26:40 AM MDT

Avaya Aura® Experience Portal 6.0 (ExperiencePortal) Home Help Logoff

Expand All | Collapse All

- ▼ **User Management**
 - Roles
 - Users
 - Login Options
- ▼ **Real-Time Monitoring**
 - System Monitor
 - Active Calls
 - Port Distribution
- ▼ **System Maintenance**
 - Audit Log Viewer
 - Trace Viewer
 - Log Viewer
 - Alarm Manager
- ▼ **System Management**
 - Application Server
 - MPP Manager
 - Software Upgrade
 - System Backup
- ▼ **System Configuration**
 - Alarm Codes
 - Alarm/Log Options
 - Applications
 - EPM Servers
 - MPP Servers
 - Report Data
 - SNMP
 - Speech Servers
 - VoIP Connections
- ▼ **Security**
 - Certificates
 - Licensing
- ▼ **Reports**
 - Standard
 - Custom
 - Scheduled
- ▼ **POM**
 - POM Home
 - POM Monitor

You are here: Home

Avaya Aura® Experience Portal Manager

Avaya Aura® Experience Portal Manager (EPM) is the consolidated web-based application for administering Experience Portal. Through the EPM interface, you can configure Experience Portal, check the status of a Experience Portal component, and generate reports related to system operation.

Installed Components

Media Processing Platform
Media Processing Platform (MPP) is an Avaya media processing server. When an MPP receives a call from a PBX, it invokes a VoiceXML or CCXML application on an application server and communicates with ASR and TTS servers as necessary to process the call.

Proactive Outreach Manager
Avaya Proactive Outreach Manager (POM) provides a solution for unified, multichannel, inbound and outbound architecture, with the capability to communicate through different channels of interaction, from Short Message Service (SMS) to e-mail to the traditional voice and video.

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5.1. Add VoIP Connections

During compliance testing both H.323 and SIP VoIP connections were used. However, either H.323 or SIP VoIP connection can be active at any given time. On the left pane, click on the **System Configuration → VoIP Connections** tab to configure VoIP connections (not shown).

5.1.1. H.323 Connection

To add an H.323 Connection, click on **H.323** tab (not shown) and click **Add** (not shown)

- **Name:** Enter a descriptive name
- **Gatekeeper Address:** Enter the IP address of Communication Manager.
- **Media Encryption:** Set to **No**.
- **New Stations:** Enter **Station From** and **To**, and **Password**. Select **Inbound and Outbound** and click **Add**.
Note: Station information should be gathered from the existing Communication Manager configuration.
- Retain the default values in the remaining fields Click **Save** to save changes.

Welcome, admin
Last logged in today at 11:26:40 AM MDT

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)
Home Help Logoff

Expand All Collapse All

▼ User Management
Roles
Users
Login Options

▼ Real-Time Monitoring
System Monitor
Active Calls
Port Distribution

▼ System Maintenance
Audit Log Viewer
Trace Viewer
Log Viewer
Alarm Manager

▼ System Management
Application Server
MPP Manager
Software Upgrade
System Backup

▼ System Configuration
Alarm Codes
Alarm/Log Options
Applications
EPM Servers
MPP Servers
Report Data
SNMP
Speech Servers
VoIP Connections

▼ Security
Certificates
Licensing

▼ Reports
Standard
Custom
Scheduled

▼ POM
POM Home
POM Monitor

You are here: Home > System Configuration > VoIP Connections > Add H.323 Connection

Add H.323 Connection

Use this page to add a new H.323 connection.

Name: CM5052

Enable: ☒ Yes ☐ No

Gatekeeper Address: 10.64.50.52

Alternative Gatekeeper Address:

Gatekeeper Port: 1719

Media Encryption: ☐ Yes ☒ No

New Stations

	From	To
Station:	60001	60002
Password:	••••••	
	<input checked="" type="radio"/> Same Password <input type="radio"/> Use sequential passwords	
Station Type:	Inbound and Outbound Inbound Only Maintenance	

Add

Configured Stations (M for Maintenance, I for Inbound Only)

<No Station>

Remove

Save

Cancel

Help

5.1.2. SIP Connection

To add a **SIP Connection**, click on the **SIP** tab (not shown) on the **VoIP Connections** page (not shown).

- **Name:** Enter a descriptive name..
- Set **Proxy Transport** to **TCP**.
- In the **Address** and **Port** boxes, enter the IP address and Port of Session Manager.
- **SIP Domain:** Enter the domain used in Session Manager.
- **Maximum Simultaneous Calls:** During the test, **10** was used for the Maximum Simulataneous Calls field.
- Retain the default values in the remaining fields Click **Save** to save changes.

Welcome, admin
Last logged in today at 11:26:40 AM MDT

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

Home
Help
Logout

Expand All | Collapse All

User Management
Roles
Users
Login Options

Real-Time Monitoring
System Monitor
Active Calls
Port Distribution

System Maintenance
Audit Log Viewer
Trace Viewer
Log Viewer
Alarm Manager

System Management
Application Server
MPP Manager
Software Upgrade
System Backup

System Configuration
Alarm Codes
Alarm/Log Options
Applications
EPM Servers
MPP Servers
Report Data
SNMP
Speech Servers
VoIP Connections

Security
Certificates
Licensing

Reports
Standard
Custom
Scheduled

POM
POM Home
POM Monitor

You are here: [Home](#) > System Configuration > [VoIP Connections](#) > Add SIP Connection

Add SIP Connection

Use this page to add a new SIP connection.

Name:

Enable: ☒ Yes ☐ No

Proxy Transport:

☒ Proxy Servers ☐ DNS SRV Domain

Address	Port	Priority	Weight	
10.64.50.31	5060	0	0	Remove

[Additional Proxy Server](#)

Listener Port:

SIP Domain:

P-Asserted-Identity:

Maximum Redirection Attempts:

Consultative Transfer: ☒ INVITE with REPLACES ☐ REFER

SIP Timers

T1: millisecond(s)

T2: millisecond(s)

B and F: millisecond(s)

Call Capacity

Maximum Simultaneous Calls:

☒ All Calls can be either inbound or outbound

☐ Configure number of inbound and outbound calls allowed

5.2. Add Speech Servers

On the left pane, click on the **System Configuration** → **Speech Servers** tab to add Speech Server.

5.2.1. ASR Server

To add an **ASR** server, click on **ASR** tab (not shown), and click **Add** (not shown).

- **Name:** Enter a descriptive name.
- **Enable:** Set to **Yes**.
- **Engine Type:** Set to **Nuance**, using the drop down menu
- **Network Address:** Enter the IP address of LumenVox Automated Speech Recognizer.
- **Base Port:** Enter **554**.
- **Total Number of Licensed ASR Resources:** Enter an appropriate value
- **New Connection per Session:** Select **Yes**.
- **Languages:** Select **English(USA) en-US**.

- **RTSP URL:** Enter<LumenVox_ASR_IP address>/media/speechrecognizer
- Click **Save** to save changes.

Welcome, admin
Last logged in today at 11:26:40 AM MDT

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

Home
Help
Logoff

Expand All | Collapse All

▼ **User Management**
Roles
Users
Login Options

▼ **Real-Time Monitoring**
System Monitor
Active Calls
Port Distribution

▼ **System Maintenance**
Audit Log Viewer
Trace Viewer
Log Viewer
Alarm Manager

▼ **System Management**
Application Server
MPP Manager
Software Upgrade
System Backup

▼ **System Configuration**
Alarm Codes
Alarm/Log Options
Applications
EPM Servers
MPP Servers
Report Data
SNMP
Speech Servers
VoIP Connections

▼ **Security**
Certificates
Licensing

▼ **Reports**
Standard
Custom
Scheduled

▼ **POM**
POM Home
POM Monitor

You are here: [Home](#) > System Configuration > [Speech Servers](#) > Add ASR Server

Add ASR Server

Use this page to configure Experience Portal to communicate with a new ASR server.

Name:

Enable: ☒ Yes ☐ No

Engine Type:

Network Address:

Base Port:

Total Number of Licensed ASR Resources:

New Connection per Session: ☒ Yes ☐ No

Languages:

Dutch(Netherlands) nl-NL
English(Australia) en-AU
English(UK) en-GB
English(India) en-IN
English(Singapore) en-SG
English(USA) en-US

MRCP

Ping Interval: second(s)

Response Timeout: second(s)

Protocol:

RTSP URL:

Save

Cancel

Help

5.2.2. TTS Server

To add a TTS server, click on **TTS** tab (not shown) on **Speech Servers** (not shown) page, and click **Add** (not shown).

- **Name:** Enter a descriptive name.
- **Enable:** Set to **Yes**.
- **Engine Type:** Set to **Nuance** using the drop down menu.
- **Network Address:** Enter the IP address of LumenVox Text to Speech server.
- **Base Port:** Enter **554**.
- **Total Number of Licensed TTS Resources:** Enter an appropriate value.
- **New Connection per Session:** Select **Yes**.
- **Languages:** Select **English(USA) en-US Jennifer F**.
- **RTSP URL:** Enter **<LumenVox_ASR_IP address>/media/speechsynthesizer**
- Click **Save** to save changes.

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Avaya Aura® Experience Portal 6.0 (ExperiencePortal) Home Help Logoff

Expand All | Collapse All

▼ User Management
Roles
Users
Login Options

▼ Real-Time Monitoring
System Monitor
Active Calls
Port Distribution

▼ System Maintenance
Audit Log Viewer
Trace Viewer
Log Viewer
Alarm Manager

▼ System Management
Application Server
MPP Manager
Software Upgrade
System Backup

▼ System Configuration
Alarm Codes
Alarm/Log Options
Applications
EPM Servers
MPP Servers
Report Data
SNMP
Speech Servers
VoIP Connections

▼ Security
Certificates
Licensing

▼ Reports
Standard
Custom
Scheduled

▼ POM
POM Home
POM Monitor

You are here: [Home](#) > [System Configuration](#) > [Speech Servers](#) > Add TTS Server

Add TTS Server

Use this page to configure Experience Portal to communicate with a new TTS server.

Name:

Enable: ☒ Yes ☐ No

Engine Type:

Network Address:

Base Port:

Total Number of Licensed TTS Resources:

New Connection per Session: ☐ Yes ☒ No

Voices:

MRCP

Ping Interval: second(s)

Response Timeout: second(s)

Protocol:

RTSP URL:

Save **Cancel** **Help**

6. Configure LumenVox Automated Speech Recognizer

All configurations for LumenVox applications were performed by a LumenVox Engineer.

Log on to LumenVox server using a SSH client. The `/etc/lumenvox/media_server.conf` file needs to be modified for the following fields:

- The value of **mrsp_server_ip** must be set to the IP address of the machine that LumenVox is installed on. This must be an IP address that the Experience Portal can reach and route traffic to. Please contact LumenVox Support for questions about configuring firewalls if they will be running.
- The value of **compatibility_mode** must be changed from the default **0** to **1**

Note: When configuring an application in Experience portal to use the LumenVox ASR, set the "Speech Complete Timeout" parameter under **Speech Parameters** to a non-0 value:

Speech Parameters ▾

ASR

Confidence Threshold:

Sensitivity Level:

Speed vs. Accuracy:

N Best List Length:

No Input Timeout:

millisecond(s)

Recognition Timeout:

millisecond(s)

Speech Complete Timeout:

800

millisecond(s)

Speech Incomplete Timeout:

millisecond(s)

Maximum Grammar Cache Age:

second(s)

Minimum Grammar Freshness Time:

second(s)

Maximum Grammar Staleness:

second(s)

Vendor Parameters:

TTS

Prosody Volume:

<NONE> ▾

or

Prosody Rate:

<NONE> ▾

or

Vendor Parameters:

7. Configure LumenVox Text-to-Speech Server

The LumenVox Media Server must be configured as described in **Section 6**. There are no special configurations for the Text-to-Speech Server.

8. Verification Steps

This section provides the verification steps that may be performed to verify that Experience Portal and the LumenVox servers are online and functioning properly.

8.1. Avaya Aura® Experience Portal

- From the EPM web interface, navigate to **System Management → MPP Manager**. From the MPP Manager screen, shown below, verify that the Media Processing Platform (MPP) servers are **Online** and **Running**.

Welcome, admin

Last logged in today at 2:18:12 PM MDT

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

[Home](#)
[? Help](#)
[Logoff](#)

Expand All | Collapse All

▼ User Management

Roles

Users

Login Options

▼ Real-Time Monitoring

System Monitor

Active Calls

Port Distribution

▼ System Maintenance

Audit Log Viewer

Trace Viewer

Log Viewer

Alarm Manager

▼ System Management

Application Server

MPP Manager

Software Upgrade

System Backup

▼ System Configuration

Alarm Codes

Alarm/Log Options

Applications

EPM Servers

MPP Servers

Report Data

SNMP

Speech Servers

VoIP Connections

▼ Security

Certificates

Licensing

▼ Reports

Standard

Custom

Scheduled

▼ POM

POM Home

POM Monitor

You are here: [Home](#) > System Management > MPP Manager

MPP Manager (Jun 26, 2013 2:21:31 PM MDT)

[Refresh](#)

This page displays the current state of each MPP in the Experience Portal system. To enable the state and mode commands, select one or more MPPs. To enable the mode commands, the selected MPPs must also be stopped.

Last Poll: Jun 26, 2013 2:21:20 PM MDT

	Server Name	Mode	State	Config	Auto Restart	Restart Schedule		Active Calls	
						Today	Recurring	In	Out
<input type="checkbox"/>	MPPRemote	Online	Running	OK	No	No	None	0	0

State Commands

Start

Stop

Restart

Reboot

Halt

Cancel

Mode Commands

Offline

Test

Online

Restart/Reboot Options

☐ One server at a time
☒ All selected servers at the same time

Help

- On the left pane, navigate to **Real-Time Monitoring → Port Distribution**. From the Port Distribution page, verify that the ports on the MPP server are in service

Welcome, admin
Last logged in today at 2:18:12 PM MDT

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

Home
Help
Logoff

Expand All | Collapse All

▼ **User Management**
Roles
Users
Login Options

▼ **Real-Time Monitoring**
System Monitor
Active Calls
Port Distribution

▼ **System Maintenance**
Audit Log Viewer
Trace Viewer
Log Viewer
Alarm Manager

▼ **System Management**
Application Server
MPP Manager
Software Upgrade
System Backup

▼ **System Configuration**
Alarm Codes
Alarm/Log Options
Applications
EPM Servers
MPP Servers
Report Data
SNMP
Speech Servers
VoIP Connections

▼ **Security**
Certificates
Licensing

▼ **Reports**
Standard
Custom
Scheduled

▼ **POM**
POM Home
POM Monitor

You are here: [Home](#) > Real-Time Monitoring > Port Distribution

Port Distribution (Jun 26, 2013 2:22:34 PM MDT)

Refresh

This page displays information about how the telephony resources have been distributed to the MPPs. You configure the telephony resources on the VoIP Connections page.

Total Ports: 14

Last Poll: Jun 26, 2013 2:22:21 PM MDT

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
50011	Online	In service	CM5052	H323	MPPRemote	
55555	Online	In service	CM5052	H323	MPPRemote	
25508	Online	In service	Jacada	H323	MPPRemote	
25509	Online	In service	Jacada	H323	MPPRemote	
1	Online	In service	sm5031	SIP_Trunk	MPPRemote	
2	Online	In service	sm5031	SIP_Trunk	MPPRemote	
3	Online	In service	sm5031	SIP_Trunk	MPPRemote	
4	Online	In service	sm5031	SIP_Trunk	MPPRemote	
5	Online	In service	sm5031	SIP_Trunk	MPPRemote	
6	Online	In service	sm5031	SIP_Trunk	MPPRemote	
7	Online	In service	sm5031	SIP_Trunk	MPPRemote	
8	Online	In service	sm5031	SIP_Trunk	MPPRemote	
9	Online	In service	sm5031	SIP_Trunk	MPPRemote	
10	Online	In service	sm5031	SIP_Trunk	MPPRemote	

Help

8.2. LumenVox Automated Speech Recognizer

The Avaya test application (usually installed with the MPP in `/mpp/misc/avptestapp/intro.vxml`) may be used to test the ASR.

8.2.1. Configuring the Application

From the EPM web interface, navigate to **System Configuration → Applications**. Click the **Add** button (not shown) to create a new application.

- **Name:** Enter a descriptive name.
- **Enable:** Set to **Yes**.
- **Type:** Set to **VoiceXML** using the drop down menu.
- **URL** Enter the URL for the Application.
- **Speech Servers** Select **Nuance** for **ASR** and **TTS**.
- **Application Launch** Enter an Extension for **Called Number:** and click the **Add** button.
- **Speech Parameters** Enter **800** for **Speech Complete Timeout:**
- Click **Save** to save changes.

Note: When configuring an application in Experience portal to use the LumenVox ASR, set the "**Speech Complete Timeout**" parameter under **Speech Parameters** to a non-0 value:

Welcome, admin
Last logged in today at 9:49:32 AM MDT

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)
Home Help Logoff

Expand All Collapse All

User Management

Roles
Users
Login Options

Real-Time Monitoring

System Monitor
Active Calls
Port Distribution

System Maintenance

Audit Log Viewer
Trace Viewer
Log Viewer
Alarm Manager

System Management

Application Server
MPP Manager
Software Upgrade
System Backup

System Configuration

Alarm Codes
Alarm/Log Options
Applications
EPM Servers
MPP Servers
Report Data
SIMP
Speech Servers
VoIP Connections

Security

Certificates
Licensing

Reports

Standard
Custom

POM

POM Home
POM Monitor

You are here: Home > System Configuration > Applications > Add Application

Add Application

Use this page to deploy and configure a new application on the Experience Portal system.

Name:

Enable: ☒ Yes ☐ No

Type:

URI

☒ Single
☐ Fail Over
☐ Load Balance

VoiceXML URL:

Mutuel Certificate Authentication: ☐ Yes ☒ No

Basic Authentication: ☐ Yes ☒ No

Speech Servers

ASR: TTS:

Languages:
Voices:

Application Launch

☒ Inbound
☐ Inbound Default
☐ Outbound

☒ Number
☐ Number Range
☐ URI

Called Number:

Speech Parameters

ASR

Confidence Threshold:
Sensitivity Level:
Speed vs. Accuracy:
N Best List Length:
No Input Timeout: millisecond(s)
Recognition Timeout: millisecond(s)
Speech Complete Timeout: millisecond(s)
Speech Incomplete Timeout: millisecond(s)
Maximum Grammar Cache Age: second(s)
Minimum Grammar Freshness Time: second(s)
Maximum Grammar Staleness: second(s)

Vendor Parameters:

TTS

Prosody Volume: or
Prosody Rate: or

Vendor Parameters:

Reporting Parameters

Advanced Parameters

8.2.2. Verifying the LumenVox ASR

- Dial into the application using the Called Number specified in the newly created application.
- At the main menu, press 1 for speech recognition test.
- When prompted, speak "Open the window."
- Confirm that the application understands the utterance.

Note: For optimal results, avoid use of a speakerphone when testing the ASR, as it may introduce recognition issues.

8.3. LumenVox Test-to-Speech Server

The Avaya test application (usually installed with the MPP in `/mpp/misc/avptestapp/intro.vxml`) may be used to test the TTS. Perform the following steps:

1. Configure the test application to use the LumenVox TTS.
2. Dial into the application.
3. At the main menu, press 2 for text-to-speech test.
4. Confirm TTS speaking.

9. Conclusion

These Application Notes describe the configuration steps required to integrate LumenVox Automated Speech Recognizer, LumenVox Text-to-Speech Server, LumenVox Call Progress Analysis with Avaya Aura® Experience Portal. All feature and serviceability test cases were completed successfully without any observations as described in **Section 2.2**.

10. Additional References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at <http://support.avaya.com>.

[1] *Administering Avaya Aura® Experience Portal*, June 2013

LumenVox help documentation, including detailed installation and configuration instructions, is available online at <http://www.lumenvox.com/help/>

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